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Robust Active Noise Control System for Fighter Aircraft Pilot Helmet Application

Y. K. Bharath[a,*], G. Chitralekha[c], S. Veena[b], H. Lokesha[b], K. V. Nagalakshmi[a] and U. Dilna[c]

[a]*National Institute of Engineering, Manandavadi Road, Mysuru 570 008, India*
[b]*National Aerospace Laboratories, Old Airport Road, Bengaluru 560 017, India*
[c]*Reva Institute of Technology and Management, Kattigenahalli, Bengaluru 560 064, India*

Abstract

This paper proposes an Active Noise Control (ANC) scheme for fighter aircraft pilot helmet application. The proposed scheme addresses the noise environment inside the helmet to achieve perceivable noise attenuation. It also incorporates algorithms such as energy based detectors to control the operation of ANC system and variable step-size to increase the performance of ANC, to make the system robust. The paper highlights the real-time algorithm development on a DSP processor to meet the real-time resource constraints. The developed ANC system is evaluated in the laboratory for a typical fighter aircraft noise and the results when compared with the performance of existing system, found to be quite promising.

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Keywords: Adaptive Noise Canceller; Fighter Aircraft Pilot Helmet; FXLMS; LMS; Robust Active Noise Control; Variable Step-size.

1. Introduction

The cockpit of a fighter aircraft will be very noisy and their mean levels range from 95 to 105 dB. This exceeds the damage risk criterion of 8 h/day exposure. Hence, may result in hearing impairment to the pilot, especially during prolonged exposures. Increased periodontal disease in aircrew members and cardiovascular risks demands the reduction of noise levels. Further, speech intelligibility and recognition of warning signals is adversely affected in a noisy environment as the primary effect of noise is masking of voice communication signals.

The prime purpose of a fighter aircraft pilot helmet is protection and it is made of high strength, lightweight para-armid reinforced with epoxy resin shell and the helmet earcups use open-cell foam rubber. In addition to protection, it filters out most of the noise signals of high frequency range, hence providing passive attenuation. Owing to broadband nature of the noise, considerable low frequency noise will be audible in the earcup. This can be addressed using ANC technology.

*Corresponding author. Tel.: +91 9035118116.
E-mail address: bharath99yk@gmail.com

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Considerable developmental effort has been spent on Active Noise Reduction (ANR) headsets[1] and are being successfully used in aircraft cabin environments, where the ambient noise level is very low compared to fighter aircraft cockpits. Few of the ANR helmets availableare based on passive electronic components or feedback control principles[2]. The use of passive electronic components lacks the ability to adapt to changing noise environments and its performance is very poor for broadband noise. Alternatively, the use of feedback algorithm (FBANC), which cannot differentiate between noise and communication signals, may affect the speech quality. Hence, there is a necessity to clearly understand the requirements to develop a helmet ANC system.

The noise perceived inside the helmet earcup is due to two components[3]. First one is the noise component that enters directly through the helmet, referred as Cockpit Noise Direct (CND) and second one is the noise component that enters through the pilot microphone and Audio Management Unit (AMU), referred as Cockpit Noise AMU (CNA). The presently existing Helmet ANC systems only address CND. To have an effective perceived noise reduction, it is necessary to address both these noise components.

The proposed method addresses CND, which is realized using Feed Forward approach (FFANC) proposed by Sen and Morgan[4], this ensures that the communication signals are not affected by ANC operation. For catering CNA, an adaptive noise canceller is used, which uses Least Mean Square (LMS) algorithm[5].

The ANC system might diverge in case of absense or due to sudden variations of the cockpit noise. The paper tries to address this by incorporating energy-based conditions[6] to limit the ANC residual error to pre-defined bounds. A good ANC system is expected to have faster convergence rate and lower residual error. Therefore, it is proposed to incorporate Variable Step-Size (VSS) algorithm[7] in this paper.

Further, the above mentioned algorithms were developed on Texas Instruments' TMS320C6748 floating point DSP processor. As far as the real-time performance of ANC is concerned, increasing sampling rate of the system extends the frequency range of ANC operation and resulting in better attenuation. However the real-time coding using a high-level language is unable to achieve desirable performance. Hence, the development of ANC algorithms using processor specific assembly language is considered in this paper.

Nomenclature

ANC	Active Noise Control
AMU	Audio Management Unit
CND	Cockpit Noise that enters pilot helmet Directly
CNA	Cockpit Noise that enters pilot helmet through AMU
FXLMS	Filtered Least Mean Square
FFANC	Feedforward Active Noise Control
STE	Short-Term Energy
NSS	Normalized Step-Size
VSS	Variable Step-Size

2. Algorithms for Helmet ANC System

This section gives details of the algorithms used in realizing helmet ANC. These are categorized under algorithms for noise attenuation and algorithms for incorporating robustness into the system. Each of them is described in the sections below.

2.1 Algorithms for noise attenuation

Figure 1 shows the block schematic of the proposed helmet ANC system. As mentioned earlier, there are two components of noise CND and CNA, to be addressed to achieve good attenuation inside the helmet earcups. CND is addressed by FFANC and CNA by the adaptive noise canceller algorithms. Both of them use the same cockpit noise as reference. This noise reference $x(n)$ is obtained using a microphone, which is placed inside the cockpit.

Two separate channels of FFANC are used for each earcups. The generated anti-noise is fed to the loudspeakers present in the earcups. An error microphone placed inside each of the earcup gives the residual noise signals.

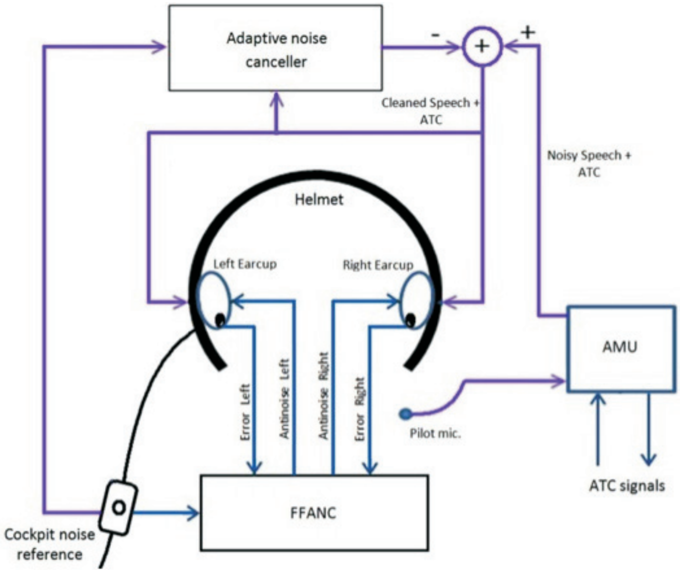

Fig. 1. Schematic of Noise Attenuation Scheme for Fighter Aircraft Pilot Helmet.

As shown in Fig. 1, the path between the reference microphone and the loudspeaker is known as main path. Identification of this path is necessary for suitable generation of anti-noise. To accomplish this, Filtered Input LMS (FXLMS) algorithm is used in which, the adaptation of weights of main path filter is done using the filtered input signal $x'(n)$ and error signal $e(n)$ as shown in Eq. (1).

$$W_k(n+1) = W_k(n) + \mu(n)e(n)x'(n-k) \tag{1}$$

where,

$W_k(n)$ - adaptive filter with L taps.
$e(n)$ - error microphone pickup after injecting anti-noise, where the anti-noise is given by Eq. (2).

$$y(n) = \sum_{k=0}^{L-1} W_k(n)x(n-k) \tag{2}$$

$x'(n)$ - Input filtered with secondary path $S(z)$ and is given by Eq. (3).

$$x'(n) = S_k(n) * x(n) \tag{3}$$

$S(z)$ represents the physical path between loudspeaker and error microphone, which is known as secondary path and it plays a very crucial role in the convergence of ANC. The path $S(z)$ is estimated offline by additive random noise technique, with an adaptive filter of M taps, using LMS algorithm.

The FXLMS algorithm as explained above is sensitive to the $x(n)$ scaling of its input. This makes it very hard to choose a learning rate μ that guarantees stability of the algorithm. Hence, to solve this problem, Normalizing the Step-Size (NSS) with power of the input $P(n)$ is considered, which is done as shown in Eq. (4) and (5).

$$P(n+1) = \beta_{ns}P(n) + (1-\beta_{ns})x'(n)^2 \tag{4}$$

$$\mu(n+1) = \frac{\mu(n)}{(L+M)\times P(n+1)} \tag{5}$$

To address CNA, as mentioned earlier an adaptive noise canceller is used in this paper. Here, the path between the reference microphone and pilot microphone is estimated using LMS algorithm, in which the weights of the filter are updated according to Eq. (6). The estimated signal, $est(n)$ is calculated using Eq. (7). The residual signal is the noise free speech, $sp(n)$ calculated using Eq. (8), and is added to the anti-noise and fed to the loudspeakers as shown in Fig. 1.

$$W_k(n+1) = W_k(n) + \mu(n)sp(n)x(n-k) \tag{6}$$

$$est(n) = \sum_{k=0}^{L-1} W_k(n)x(n-k) \tag{7}$$

$$sp(n) = \text{AMU signal} - est(n) \tag{8}$$

2.2 Algorithms to increase Robustness of ANC

In this section, the algorithms that are incorporated in the system to make robust ANC; a real-time application, are discussed.

2.2.1 Noise sensitive algorithms for ANC

ANC unit is an embedded system and hence the ANC code resides in the flash memory of the DSP present in the system. The anti-noise generation is based on preset parameters meant to meet the cockpit noise level. Once power ON, the system starts producing anti-noise based on the signal picked up from reference microphone, irrespective of presence of noise in the cockpit. In case of absence of noise, this leads to unpleasant sound inside the earcups. Hence to turn ON the ANC only in the presence of cockpit noise, an energy-based measure is used. Here, Short Term Energy (STE) feature of the reference noise signal is considered to differentiate between noise and silence.

$$\text{STE of ref signal} = \sum_{n=1}^{L} x(n)^2 \tag{9}$$

Reference microphone signal is picked up in the absence of noise for a time duration and its energy is calculated. Based on this a threshold is set. The ANC system is turned ON, only if the STE of reference noise signal calculated using Eq. (9), crosses the threshold.

$$\text{STE of error signal} = \sum_{n=1}^{L} e(n)^2 \tag{10}$$

Further, ANC system may diverge in case of some sudden variations of the cockpit noise and result in large amplitude of unpleasant noise in the earcups. To prevent this, the STE of the error signal is taken into account. Here, the energy of the error signal in the presence of noise with ANC OFF is used to calculate a threshold. If the STE of error microphone signal, which is calculated using Eq. (10), crosses the threshold, the weights of the mainpath filter W_k are reset so that the anti-noise starts to build from the initial point.

2.2.2 Variable step-size (VSS) algorithm for ANC

In LMS/FXLMS algorithm, step-size parameter μ decides the convergence speed and final residual error level. Customarily, power normalization (NSS) is used to adjust the value of step-size in the algorithm. Since residual error is audible to pilot's ears, it has to be very low and also it is desirable to have faster error convergence rate. However with a fixed μ, any one of these is achievable. To achieve both, the computationally efficient robust Variable Step-Size (VSS) algorithm is applied as given below,

$$G_k(n+1) = \beta_{vs}\, G_k(n) + (1-\beta_{vs})e(n)x'(n-k) \tag{11}$$

$$\mu(n+1) = \alpha\mu(n) + \sigma \sum_{k=0}^{L-1} G_k^2(n) \tag{12}$$

The VSS alorithm estimates both gradient $G_k(n)$ and step-size $\mu(n)$ as shown in Eq. (11) and (12), based on Griffiths' cross crorrelation between the filtered input and error. This enables to have a larger step-size intially, to have a faster convergence and reduced step-size towards convergence to achieve a smaller convergence error.

2.2.3 Choosing assembly language for algorithms used in ANC

The operating frequency range of ANC is decided by the speed with which ANC algorithm gets excecuted. For example, if a code to generate 1 sample of ant-noise takes 1ms, then the sampling frequency required is 1 KHz and this limits the ANC operating range to 500 Hz. Apart from FFANC and adaptive noise canceller, inclusion of robust features as explained earlier in this section calls for higher sampling rate. Also, it is observed that the perception of noise reduction will be better when the sampling rate is incresed for processing.

In comparison to C code, the assembly code can achieve higher sampling frequencies as they are designed to exploit the processor architecture. In view of this, the coding of the algorithms in assembly language has contributed to the robustness of the ANC system.

3. Helmet ANC System

This section describes the helmet ANC system developed at CSIR-NAL.

The components of the system as shown in Fig. 2 are,

- Controller – The heart of this unit is a TMS320C6748 board with ADCs and DACs for the algorithm development. The signal conditioner board comprises of power amplifiers, pre amplifiers, filters and circuitry, a signal recording facility to capture the earcup noise signals for off-line analysis. The unit is battery operated and has a backup of about 2 hours. This controller unit is compact enough to be fitted into pilot's suit pocket.
- Transducers – The same loudspeakers present in the helmet earcups for communication purpose are used to give out the anti-noise and Electret Condenser microphones are used for picking reference and error signals.
- Pilot helmet – The ANC unit and typical fighter aircraft helmet integrated with transducers are as shown in Fig. 2.

Fig. 2. The ANC Unit Integrated with the Fighter Aircraft Pilot Helmet.

Table 1. Parameters for FFANC using FXLMS Without VSS.

Parameter	μ_r	μ_l	β_{nr}	β_{nl}
Value	0.1	0.1	0.999	0.999

Table 2. Parameters for Adaptive Noise Canceller using LMS.

Parameter	μ	β
Value	0.1	0.999

Table 3. Parameters for FFANC using FXLMS with VSS.

Parameter	μ_r max	μ_r min	μ_l max	μ_l min	β_{nr}	β_{nl}
Value	1	0.1	1	0.1	0.999	0.999
Parameter	β_{vr}	β_{vl}	α_r	α_l	σ_r	σ_l
Value	0.999	0.999	0.0001	0.0001	0.99	0.99

4. Helmet ANC System Evaluation and Results

Initially the algorithms were developed in MATLAB and a detailed analysis was carried out to assess their suitability to helmet ANC application in terms of performance and execution time involved. Later, they were developed Embedded-C and assembly programming languages for TMS320C6748 processor.

The parameters set for ANC algorithmare tabulated in Table 1, 2 and 3 as shown. The subscripts r and l in the parameter list indicates the right and left channels of the ANC, $n$and v refers for the normalizing and variable step-size respectively.

4.1 System evaluation using MATLAB

The ANC system including CND-CNA loops was developed using MATLAB[8,9]. The main path and secondary path are modeled to be in the form of impulse response[4] and recorded fighter aircraft noise was used for simulation.

Both FFANC and adaptive noise canceller loops having parameters shown in Table 1–3, performed as expected. Those results are not presented here, as the noise reduction achieved in case of MATLAB simulations will not reveal any meaningful insight towards what can be achieved in real-time. The results presented in this section justify the applicability of algorithms introduced to bring in robustness.

4.1.1 Evaluation of variable step-size algorithm

The FFANC using FXLMS algorithm with and without VSS are simulated separately in MATLAB with parameter values set according to Table 1 and 3. Simulation results of one channel are presented in Fig. 3.

The VSS algorithm applied to main path weight adaptation increases the performance of the system by increasing speed of convergence and decreasing the convergence error as indicated in Fig. 3(a). Fig. 3(b) shows the variation of step-size on convergence in accordance with Eq. (12).

4.1.2 Evaluation of noise sensitive ANC algorithms

The energy based conditions are incorporated in the ANC algorithm to detect the absence of noise and the occurrence of divergence and is simulated using MATLAB. The simulation results obtained are shown in Fig. 4.

The energy of the reference signal is calculated for a period in the absence of noise and ThA, the threshold for condition of switching ANC is set accordingly. The anti-noise is generated only when the STE of reference signal

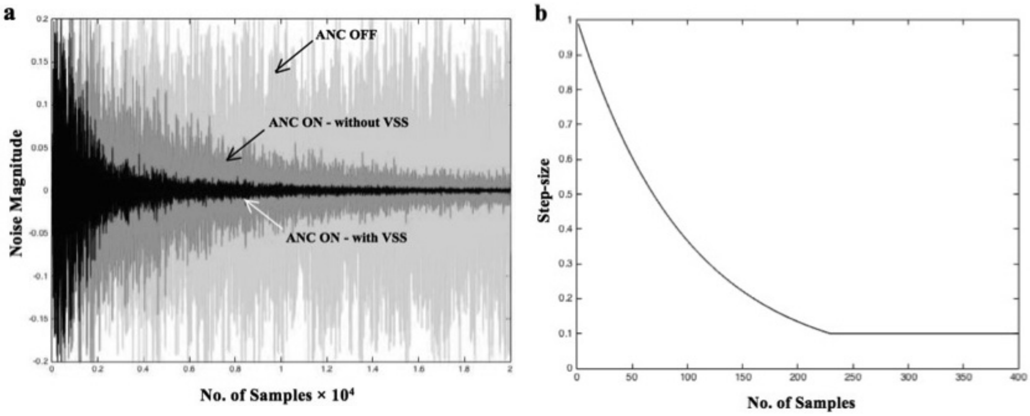

Fig. 3. (a) Convergence Error Plot with and without ANC; (b) Varying Step-size on Convergence.

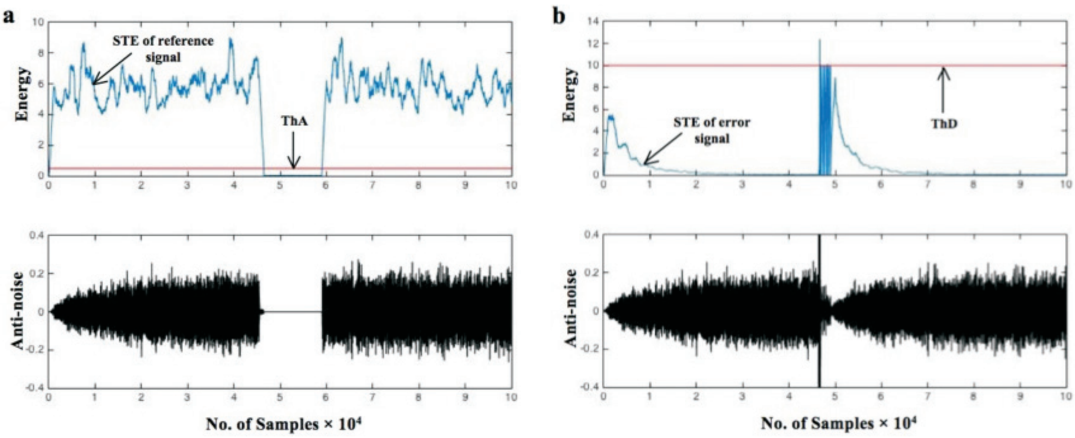

Fig. 4. (a) Switching ANC on Only in the Presence of Noise; (b) Resetting ANC on Occurrence of Divergence.

exceeds ThA as shown in Fig. 4(a). During the course of ANC ON, if the engine is turned OFF, the cockpit noise will seize to exist. For this duration, STE of reference signal will be less than ThA and in such a scenario weights of the main path filter are not adapted. The instant when the STE of signal goes high again, the adaptation of weights are resumed from where it had stopped, so that the time required for re-convergence is minimal.

The STE of the error signal is calculated during the presence of noise with ANC OFF for a period and ThD, the threshold for the condition for identifying divergence is set. Whenever the STE of the error signal exceeds ThD, the weights of the main path filter are reset and adaptation starts again from initial point as shown in Fig. 4(b).

4.2 Real-time performance evaluation of ANC system

The cockpit noise used for system evaluation was obtained from a typical fighter aircraft using M-Audio digital noise recorder and the noise levels during the flight was recorded using a B & K 2270 sound level meter. The recording obtained was of about one hour duration, which captured temporal noise during the takeoff, flight and landing conditions. From periodogram computed using MATLAB and one third octave spectrum computed using DATS analysis tool, it was concluded that the noise spectrum was of broadband nature with significant contribution due to low frequency components. The dB level recordings showed that the noise levels vary between 90 dBA–110 dBA.

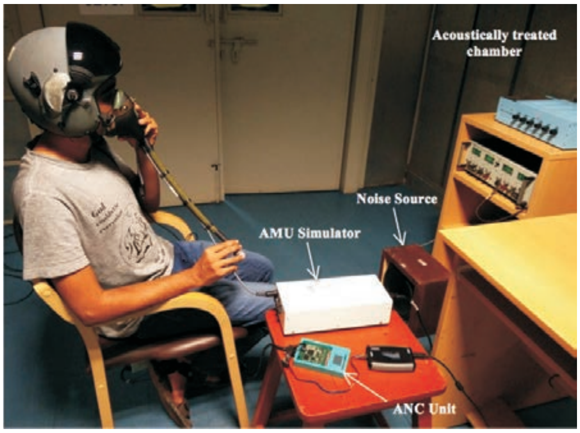

Fig. 5. Experimental Setup to Test Helmet ANC System.

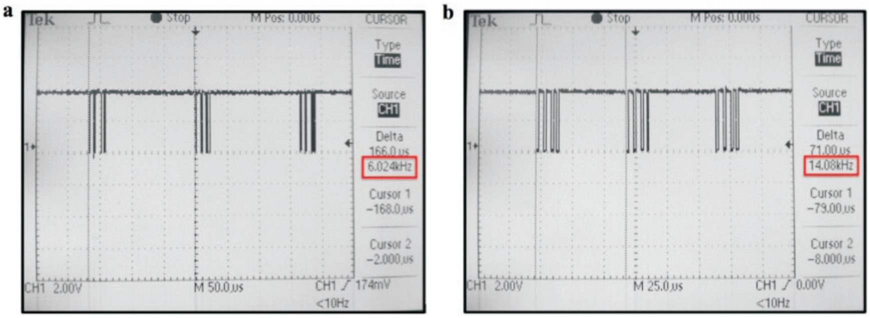

Fig. 6. (a) ADC Sampling Frequency with Embedded-C Coding; (b) ADC Sampling Frequency with Assembly Language Coding.

The experiments were conducted in an acoustically treated semi-anechoic chamber established at CSIR-NAL. The helmet is connected to the ANC unit. The pilot communication microphone output is fed to AMU simulator as shown in Fig. 5. The noise record was played using audacity software[10], which is a freeware downloadable from the Internet. The signal was fed to a 100 W loudspeaker through a power amplifier,which acts as the noise source.

4.2.1 Improvement of sampling rate with assembly programming

The existing algorithm consisting of FFANC to attenuate CND, is coded in embedded-C language[11–14] and the proposed algorithm is coded in hand optimized assembly language[11–14], their system parameters are set as per Table 1, 2 and 3. Figure 6(a) and (b) are snapshots of the oscilloscope screen measuring the sampling frequency of the ANC system achieved with C and assembly coding respectively. It was found that with C coding, the maximum achievable sampling frequency was about 6 kHz, hence the cutoff frequency of 2 kHz was selected for ANC. With assembly coding, the sampling frequency achieved was 14 kHz, hence ANC cutoff frequency was extended to 4 kHz. The increased sampling rate increases the processing speed and also enables inclusion of other algorithms required to incorporate robustness into the system.

4.2.2 Performance comparison of the existing system with proposed system

The existing ANC algorithm and proposed robust ANC algorithm are considered for system evaluation. As the unit has inbuilt storage feature, the error microphone signals with both ANC OFF and ON were stored with a sampling frequency of 16 kHz. Once the experiments were concluded, the performance of the system is analyzed using these signals. The system performance is defined by the attenuation of noise in dB, which is computed using Eq. (13).

$$\text{Attenuation in dB} = 10\log_{10}\frac{\sum_{n=1}^{N} \text{Error\_On}(n)^2}{\sum_{n=1}^{N} \text{Error\_OFF}(n)^2} \tag{13}$$

The FFT analysis was done for both the cases to verify the noise reduction in terms of individual frequency components by computing the Power Spectral Density (PSD) of the signals.

The noise attenuation in terms of decibels at particular frequencies is as shown in Fig. 7. The attenuation achieved by the existing ANC system and the proposed robust ANC system is in accordance with Eq. (13). It was observed that the ANC system handles only the noise having frequency components within the range specified by the cutoff frequency, frequencies beyond this range are handled by passive noise attenuation provided by the helmet and carry no significance with respect to ANC.

The attenuation of CND using C coded FFANC algorithm is evaluated using Eq. (13) and tabulated in Table 4. The attenuation of CND and CNA using assembly coded robust FFANC algorithm and adaptive noise canceller algorithm respectively, are individually and collectively evaluated using Eq. (13) and are also tabulated in Table 4.

It was observed from Fig. 7 and Table 4 that the performance of the proposed algorithm was superior compared to that of the existing algorithm. This was mainly due to the attenuation of CNA component, which was not addressed before. Secondly, with the use of assembly language, the sampling frequency was increased form 6 kHz to 14 kHz, because of which, the ANC operating range (cutoff frequency) was extended from 2 kHz to 4 kHz. The increased sample rate also enabled the inclusion of algorithms mentioned earlier for robustness. Further, increased sampling rate facilitated the acquisition of more noise samples per second and hence, generation of good quality anti-noise.

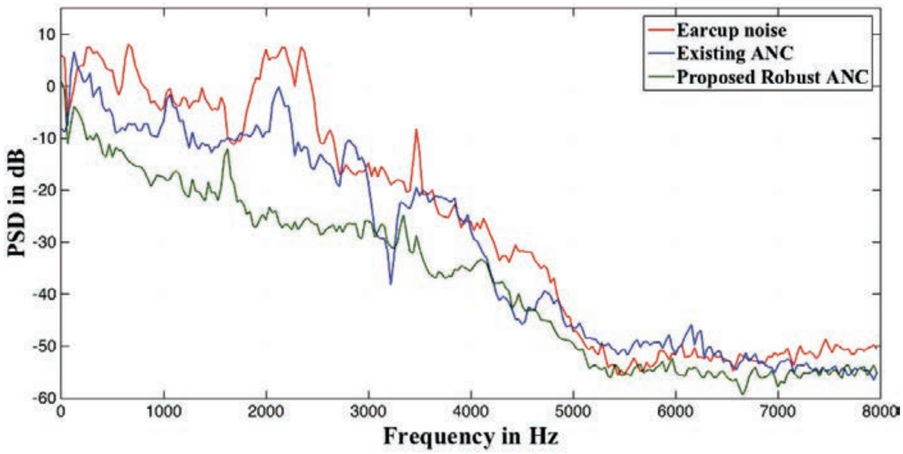

Fig. 7. Proposed Helmet ANC System Performance Depicted in Frequency Domain.

Table 4. Existing and Proposed Helmet ANC System Performances for Fighter Aircraft Cockpit Noise.

Algorithm to Attenuate	Noise Attenuation (dB) Achieved with Existing System	Noise Attenuation (dB) Achieved with Proposed System
CND	7.23	10.3
CNA	–	5.63
CND + CNA	–	16.08

5. Conclusions

This paper presents a robust ANC system suitable for fighter aircraft pilot helmet. The system addresses the noise entering earcups through direct path and AMU path. The attenuation of noise was achieved using FXLMS for FFANC and LMS algorithm for adaptive noise canceller. Further, robustness was integrated into the system by incorporating thresholds for ANC operation, VSS algorithm for fast convergence and least convergence error, and assembly language coding of algorithms for higher sampling rate. Laboratory evaluation has proved that the system is capable of the attenuating fighter aircraft cockpit noise to an appreciable extent.

Acknowledgements

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